

1 **COMBINED TELEPHONE PBX AND COMPUTER DATA ROUTER**
2 **WITH POOLED RESOURCES**
3

4 **Cross Reference to Related Applications:**

5 The present application is a continuation of application 60/125,347 filed 03/19/00
6 which was pending on the filing date of the present invention.
7

8 **Field of the invention:**

9 The present invention relates to communication systems and more particularly to PBXs
10 (private branch exchanges) for telephone circuits and to LANs (local area networks) and
11 WANs (wide area networks) for data transmission.
12

13 **Background of the Invention:**

14 Most present day offices include telephones and computers. Office suites generally
15 have a PBX or key system to handle the telephones and a separate Data Router to
16 interconnect computers via a LAN.
17

18 Telephone key systems and PBX systems are available in a wide variety of forms. Most
19 such systems are built using special purpose hardware. However, recently, programmed
20 personal computers have been used to implement telephone switching systems. Such
21 systems use the multitasking capabilities of computers such as the Microsoft Windows NT
22 system to simultaneously switch a number of telephone calls between local loops and
23 from local loops to trunk circuits.
24

1 The LANs used in office suites generally interconnect computers through Data Routers .
2 A number of companies including Cisco Systems Inc., and AT&T market Routers for
3 LAN networks.

4
5 In general, telephone systems and computer networks are moving toward similar
6 technologies. This movement has been accelerated by the advent of "Internet
7 Telephony". Internet Telephones transmit digitized and packetized voice over computer
8 LAN networks and over the Internet.

9
10 Recently, systems have become available which integrate in one unit both, telephone
11 switching or PBX capabilities, and LAN Data Routing or switching capabilities. Such
12 integrated systems can function as both a small PBX and as a LAN Data Router.

13 Presently available integrated systems use the multitasking capabilities of computer
14 systems (such as the multitasking capabilities of the Microsoft Windows NT system) to
15 handle multiple tasks on a time slice basis. Such systems can both switch telephone
16 lines and they can also route data packets traveling over a LAN network.

17
18 Recurring cost for wide area network links dominates the overall service cost when
19 providing either telephone or data communication networks. Because of this significant
20 value is provided by equipment that can achieve higher wide area network utilization
21 rates.

22
23 **Summary of the Present Invention:**

24 The present invention does not merely "integrate" the functions of a telephone switch
25 and the functions of a LAN router in one unit. The present invention "pools" resources
26 and allocates them in an optimal manner to either the task of handling telephone voice

1 traffic or to the task of handling computer data traffic. By pooling resources the present
2 invention makes optimal use of the available resources.

3
4 The present invention also provides a modular system that can be expanded easily and
5 which can meet the needs of a variety of office environments. The present invention
6 utilizes a combination of mechanisms to achieve higher statistical multiplexing on a
7 network interface by dynamically adjusting the multiplexing. The present invention:

8 1) Provides homogeneous access to DS0 trunk resource by both voice and data
9 traffic, resulting in a larger resource pool.

10 2) Partially normalizes the class of service characteristics of data traffic to make
11 it more predictable and can dynamically adjust the bandwidth such that requests
12 for resources can be honored with a higher success rate, and

13 3) Maintains multiple qualities of service for multiple voice and data streams
14 drawing from a single resource pool.

15 With the present invention data traffic is normalized into flows, each of which has an
16 assigned priority. Bandwidth is allocation to each of the flows. Filtering is performed on
17 lower priority flows if there is not sufficient bandwidth available to handle all the requests
18 for service. The bandwidth of each flow is continuously monitored and the bandwidth
19 allocation is periodically adjusted according to the assigned priority to accommodate the
20 magnitude of the requests for service and the resources available. Voice traffic is also
21 characterized as belonging to prioritized flows, though no filtering or bandwidth
22 adjustment function are applied to voice traffic because voice traffic has a constant
23 bandwidth.

24
25 Statistical multiplexing techniques have long been used in telephony networks to
26 achieve higher wide area network utilization. However, such methods have been limited

1 by several obstacles. Among the obstacles is the fact that the mechanisms used to
2 determine the maximum multiplexing rate have been limited by a common class of
3 service for all sources and destination. Another obstacle is the fact that only true
4 telephony traffic, that is voice calls, are applied to the multiplexing function. Still another
5 obstacle has been the fact that there is no ability to adapt the multiplexing rate
6 dynamically as a function of load by class of service.

7
8 The present invention achieves higher statistical multiplexing achieved by;

- 9 1) Combining different classes of service into a single, larger resource pool.
- 10 2) Dynamically adjusting both the offered load and the bandwidth available by
- 11 class of service.
- 12 3) Defining multiple {source, destination} multiplexing subgroups with different
- 13 classes of service, within the larger resource pool, to achieve different
- 14 multiplexing rates by class of service within the overall system.

15
16 **Brief Description of the Figures:**

17 Figure 1 is an overall diagram of the system.

18 Figure 2 is a programming flow diagram showing how data flows are handled in the
19 system.

20 Figure 3A is a drawing of the physical chassis that form a preferred embodiment of the
21 invention.

22 Figure 3B shows one chassis and the ports on one chassis for connection to other lines
23 or units.

24 Figure 3C is a diagram the bus arrangement which interconnect the various components
25 on chassis.

26 Figure 3D is a block diagram of one chassis.

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bandwidth numbers assigned to each flow and each sub-flow are: (Note, in the following definitions, the term flow refers to both flows and sub-flows, unless otherwise specified).

Offered Bandwidth: The bandwidth at which the system is receiving packets that as associated with a particular flow or sub-flow..

Current Bandwidth: The output bandwidth that the system has currently assigned to a particular flow. This represents the aggregate bandwidth for the DS0s that have been assigned to this flow. The Current Bandwidth will always be between the Maximum and Minimum Bandwidth.

Maximum Bandwidth: The absolute maximum bandwidth that can be assigned to a flow. This is the highest bandwidth to which "Current Bandwidth" can be set for a particular flow.

Minimum Bandwidth: The absolute minimum bandwidth that can be assigned to a flow. Minimum Bandwidth is the counterpart of Maximum Bandwidth.

Average Bandwidth: A value between Maximum Bandwidth and Minimum Bandwidth. The system attempts to maintain the Current Bandwidth for a flow equal to the Average Bandwidth by filtering as explained below. However, if the "Offered Bandwidth" exceeds the Average Bandwidth the Current Bandwidth is increased up to the Maximum Bandwidth. If the Offered Bandwidth is less than Average Bandwidth then the Current Bandwidth will be decreased down to the 'Minimum Bandwidth'.

1. The first step is to identify the problem. This involves understanding the current situation and what needs to be changed.

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17

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established at the end of each adjustment period are used for the operations that take place during the next Bandwidth Adjustment period.

The systems 1 and 2 include dial-up modems on some of the DS0 lines. Thus, some times the DS0 are used for voice; however, when the bandwidth is necessary to carry higher priority data, the line can be switched to data lines. The communication resources are therefore pooled and used for either data or telephone traffic as the demand requires. The priority assignments for data and telephones utilize the same scale, thus, the priority of communication traffic can be shared and pooled for use by the highest priority task.

The details of a system which performs the operations shown in Figure 2 will now be described. As shown in Figure 3a, the system consists of three stacked modules or chassis; however, It should be understood that the number of chassis in a stack is variable and can be selected to meet the requirements of specific application. A three-chassis stack will be used herein to illustrate the principles of the invention. As shown in Figure 3a the modules or chassis 10, 11, and 12 are interconnected by a bus 13.

The bus 13 transmits packetized information between the chassis. The information transmitted between the chassis 10, 11 and 12 includes control information, packetized voice and packetized computer data. The details of the bus which interconnects the chassis is not part of the present invention. An example of such bus is described in co-pending patent application serial number 60/098,297 filed August 27, 1998. Application serial number 60/098,297 filed August 27, 1998 is incorporated herein in its entirety by reference.

Figure 3B illustrates the connectors, which appear on each chassis 10, 11 and 12. There are six Ethernet ports 20a to 20f for connection for connection to Ethernet LAN ports such as LANs 1c1 to 1c4 shown in Figure 1. There is a connector 21 for an ISDN (Integrated Services Digital Network) line. There is a connector 22 for the ring 13 that interconnects the chassis 10, 11 and 12. There are a number of connectors 23a to 23x for connection to local loops such as telephone handsets 1t1 and 1t4 and there are a number of connectors 24a to 24x for connection to telephone trunk lines such as the T1 and DS0 lines shown in Figure 1. The number of connectors for local loops, the number of connectors for trunk lines, the number of connectors for local loops, etc. is variable depending upon the capacity desired. There is a maximum number of each that can be accommodated on a single chassis. If more than the maximum number is needed an additional chassis is used.

Figure 3C is an overall block diagram of chassis 10 which shows the RISC processor, the bus structure and the cards in the chassis. The physical structure of each chassis is conventional. The structure consists of a mother board (not explicitly shown) and plug in card designated as card A to card X. Each of the other chassis can be similar to chassis 10 or each of the chassis can have individualized characteristics by virtue of having a different set of plug in cards.

The main computational units in chassis 10 is a RISC microprocessor 305. The RISC microprocessor 305 can for example be one of the 64 bit microprocessors marketed by MIPS Corporation under the designation 46xx, 47xx or 50xx. In the specific embodiment show RISC processor 305 is a 150 Mhz IDT 4650 processor. Other models such as the 4750, 5050 or other versions may also be used. This processor family provides from 350-900 MIPS of processing power. A processor which can handle the expected traffic

1 which allocates processing power on a time slice basis to tasks which have the highest
2 priority. That is, time sensitive tasks are handled before non time sensitive tasks.

3

4 Attached to the RISC processor 305 is a DRAM memory SIMM 361 which provides up to
5 256 megabytes of random access memory. The RISC processor 305 has various
6 conventional peripheral devices such as a local flash memory 362 for fast semi-
7 permanent storage, PCMCIA Flash Card 363 for additional semi-permanent storage, a
8 Real Time Clock (RTC) 364, an Interrupt Controller, a Serial Port (UART) 365, status
9 lights and control switches 366, and a USB control port 367. All of these peripherals
10 support the 'controller' functions provided by processor 305. Processor 305 may also
11 have various other conventional peripheral devices such as terminals for administrative,
12 operational and support purposes. In addition to handling overall control of the process
13 taking place on cards a to x, the processor 305 also handles administrative support
14 functions such as maintaining the data base which keeps track of the bandwidth
15 specification for the various flows. This data can be maintained in a conventional data
16 base such as those marketed by Oracle Corporation.

17

18 The chassis can include various types of plug in card. The cards in a chassis can
19 include card A which is a combination analog trunk and analog station card, card B
20 which is a single port ISDN BRI B card and card C which is an Ethernet switching card.
21 The cards can be hot swappable and they can include cards for connection to T1/E1,
22 ISDN PRI, ADSL, T3/E3 port and others.

23

24 A bus isolation unit 351 allows RISC processor 305 to communicate with units 361 to
25 367 using bus 303 at the same time that cards A to D communicate with each other

1 using bus 303. Naturally this can not take place at a time that communication is in
2 progress between one of the cards A to D and processor 305.

3
4 Unit 352 is connected to bus 303 and to MVIP bus 316. Under control of processor 305,
5 unit 352 takes data words containing voice information from bus 303 and converts then
6 to packets on MVIP bus 316. Such a conversion is used in situation where voice data is
7 recorded (for example voice mail) in memory and it is desired to send such voice traffic
8 out on one of the voice ports such as ports 23a to 23zx. Such parallel to MVIP units are
9 conventional.

10
11 Card A includes a DSP 371-A with an associated memory SDRAM A2 which takes voice
12 traffic from station interface unit 372 and truck interface unit 373 and digitizes it for
13 transmission on MVIP bus 316. The DSP 371-A can for example be a Texas
14 Instruments model 549 DSP. Card A also includes memory SDRAM-A for transferring
15 data from RISC processor 305 to card A or for transferring information between card A
16 and the other cards via bus 303.

17
18 Card B includes a DSP 371-B with an associated memory SDRAM B2 which takes voice
19 and data traffic from ISDN interface unit 374 and either pacttizes it for transmission on
20 MVIP bus 316 or which transmits data on bus 303. . The DSP 371-B can for example be
21 a Texas Instruments model 549 DSP. Card B also includes memory SDRAM-B for
22 transferring data from RISC processor 305 to card B or for transferring information
23 between card B and the other cards via bus 303.

24
25 Card C includes and Ethernet switch 378for routing packets that LANs 20 via Ethernet
26 physical connector 379. Ethernet switch 378 has an associated Field Programmable

1 The UPC operates on 'calls' whether they are supporting voice traffic or encapsulated
2 data traffic. A "call" is herein defined as having the following states. That is, the
3 following states represent the Progression of a call:

- 4 1) signaled information and digit gathering
- 5 2) service recognition
- 6 3) address translation
- 7 4) routing
- 8 5) resource allocation

9

10 There are two basic types of calls that are handled by the system: incoming calls and
11 outgoing calls. The term 'incoming' means a call that originated outside of the system
12 and is entering the system from a trunk interface of some type. The term 'outgoing'
13 means a call that originated within the system, that is from a device directly attached or
14 within the system. Incoming calls can be destined at either a local station (trunk to
15 station) or passed on to another trunk (trunk to trunk) on the system (that is, routed
16 through the system and terminated elsewhere) or targeted at a software termination
17 point (an application within the system). Outgoing calls can be station to station or
18 station to trunk as well as being from a virtual station, that is a software termination point
19 within the system (such as the data router).

20

21 There are four types of endpoints for a call. That is, the end points can be two types of
22 trunks and two types of stations. There are physical stations and physical trunks.
23 Physical stations and physical trunks are true hardware interfaces that provide these
24 services. There are also virtual stations and virtual trunks. Virtual interfaces are
25 implemented in order to translate one service class to another. In the technical

1 literature, what are herein referred to as virtual interfaces are sometimes referred to as
2 'gateways'. Virtual stations and virtual trunks are in reality the same construct, the only
3 differentiation being that a virtual trunk is a gateway among different protocols, while a
4 virtual station only has a single termination point. For example, virtual stations to
5 implement voice mail service, while a virtual trunk can be used to implement a protocol
6 encapsulation from the data router.

7

8 To summarize, the types of calls handled by the system are:

9 Incoming, that is: a) trunk to station and b) trunk to trunk

10 Outgoing, that is: a) station to station, and b) station to trunk

11 There are two types of stations, that is: a) physical and b) virtual

12 There are two types of trunks, that is: a) physical and b) virtual

13

14 Figure 4a illustrates the progression of calls in the system. Calls enter the system as
15 either telephone calls or data calls. Block 432 represents telephone calls (voice calls)
16 entering the system, and block 431 represents data calls entering the system. (Note
17 program block 431 represents a combination of the programs 221-224, 231, 232, 233
18 and 234 which are shown in Figure 2). When a telephone call enters the system (from
19 either a local telephone port or from a WAN port), the normal signaled information and
20 dialed digits are gathered. In the service recognition phase a data base is interrogated
21 to determine the priority assigned to the particular type of call by the system
22 administrator. This is represented by block 425. Similar operations occur for new data
23 flows that are detected on one of the Ethernet ports or which enter the system via one of
24 the WAN ports.

25

1 Depending on the resources that are available and the priority of each call, the
2 telephone calls are filtered as indicated by block 435a. That is, low priority calls are not
3 assigned a DS0 circuit if there is other higher priority telephone or data traffic that
4 requires additional bandwidth. As indicated by block 435b, the data flows are filtered as
5 previously described. (note program block 435b is a combination of blocks 234 and 235).

6

7 As indicated by block 433, the resources of the system are assigned to the telephone or
8 the data traffic as dictated by the priorities.

9

10 Figure 4b illustrates Call Progression in the system. The DSP processors 371-A and
11 371-B perform the functions shown in block 401. The RISC processor 305 performs the
12 functions shown in block 402. The RISC processor has programs that do conventional
13 routing for data traffic and unified call control for voice traffic. The data router and
14 unified call control software provides data flow or call setup, not actual traffic movement.
15 Once a data call or voice call has been established, communication is directly between
16 the cards involved and the traffic does not go through the RISC processor 305.

17

18 Figure 4b shows the steps performed during the call setup progresses. As indicated by
19 blocks 405 and 406, signaling and digit gathering is done within the DSPs 371A and
20 371-B. Each station and trunk interface has a signaling path preassigned to a DSP at
21 system initialization time. The system has adequate I/O bandwidth for all stations to
22 communicate signaling bandwidth to their dedicated DSP.

23

24 As indicated by block 407, echo cancellation is provided so that the system can
25 accommodate Speech Recognition services. It is also noted that, echo cancellation is
26 widely deployed in North America and is necessary to support end-to-end circuits which

1 station call, the address translation will find the chassis and hardware address of the
2 destination address. When the service is recognized as a remote call, routing will have
3 to be completed to an appropriate trunk, before the chassis and hardware address can
4 be obtained. Once this is complete, the address translation program will then provide
5 the external addressing information. The external address information is based on both
6 the trunk type selected and the ultimate destination of the call and includes dialing
7 extensions. Such address translation is conventional.

8

9 The Client API block 411 is an interface that other services and programs in the system
10 can use to establish calls. The originating endpoint is a virtual trunk or station, as
11 described earlier. The Client API 401 has a parameter list (which is stored in memory)
12 and which specifies the service type and destination address. Each address is
13 translated, based on the service type, to account for such addressing schemes as:
14 X.121, E.164 and NSAP. In addition, specific dialing extensions can be programmable
15 for these service types, as expected.

16

17 The Client API interface is used by the Data Router to establish end to end phone calls
18 in those cases that the Data Router requires additional trunk bandwidth and needs to
19 establish additional WAN circuits to provide that bandwidth. That is the Client API
20 includes a sub-program which dials into a dial up modem to provide additional bandwidth
21 when required.

22

23 Call Routing block 412: Calls originating from a station interface may be a local station
24 to station call, or a remote station to trunk call. Remote calls have to be routed to the
25 Central Office (CO) through one of our trunk interfaces. If the call is a station to station
26 call, routing is done as follows: The called party number may be a hunt group number or

1 a direct station number. If the number is a hunt group, it is treated the same way as if it
2 was a incoming hunt group call, and apply the hunt group routing . If the number is a
3 direct station number, then it is determined if it is a station in the same chassis, or a
4 different chassis. If the call destination is on a different chassis', the call is switched over
5 the interconnect bus to the destination chassis. Note that the hook status of the
6 destination station may or may not be checked prior to this step, depending on
7 implementation considerations. In any case, a busy tone is used to indicate endpoint
8 unavailability and a reorder tone will be used to indicate resource unavailability.

9

10 As is conventional, preceding the called party number by a predetermined number,
11 usually '9' disables local PBX routing functions. Calls leaving the system can be directed
12 to a private line destined to another PBX or the Central Office. Calls destined to a
13 private line are dialed with a special preceding number, usually '8'. When calls leave the
14 system to the central office, the call is routed to outgoing trunks. The specific trunk
15 selected is determined by consulting a trunk group table.

16

17 Incoming Calls: Incoming calls enter the system through a trunk interface, that is through
18 one of the ports 24a to 24x. They may be destined to a station chassis or another trunk
19 interface. The called party may be : A direct station number (including DID), a virtual
20 station or virtual trunk number (including those connections setup by the Data Router via
21 the Client API) an ACD number, or a Hunt number).

22

23 Calls which are DID, are routed directly to the station, over the MVIP bus 316 or the
24 interconnect bus depending on the physical location of the station. A multi-stage switch
25 model is used to progress calls to foreign chassis'.

26

1 The Wide Area Network ports 24 and 31 are the only ones concerned with the pooling
2 mechanisms described below, they represent the DS0 units which are pooled and
3 allocated by these mechanisms.

4
5 WAN ports are combinations of DS0s. The breakdown is;

6 Analog trunks == 1 DS0

7 ISDN BRI == 2DS0

8 T1 == 24 DS0

9 E1 == 32 DS0

10
11 There are two basic types of communication traffic flowing through the system; a) Data
12 Traffic and b) Telephony Traffic. Data Traffic is packetized and is sent asynchronously.
13 Telephony traffic is a continuous stream and is sent synchronously.

14
15 Both types of traffic share the Wide Area Network Ports 24 and 31. In order to facilitate
16 sharing, traffic is classified , at the source, into 'flows' and traffic is assigned a bandwidth
17 specification.

18 Telephony traffic: Always uses an integral number of DS0s worth of bandwidth
19 with fixed delay and delay variation.

20 Data traffic: The system classifies data traffic into flows based on VLAN and the
21 {source, destination} IP address and {source, destination} layer four protocol port
22 pair. Based on this information the system looks up a bandwidth 'record' in
23 database 425. The offered data traffic must then be 'normalized' to the
24 'available' bandwidth through the mechanisms discussed below.

1 policing and filtering programs represented by block 435 to enforce these qualities of
2 service.

3

4 The programs discussed above and shown in block diagram form in Figure 2 are
5 implemented as programming modules in RISC processor 305. There are two
6 fundamental types of communications traffic in the system the system;

- 7 a) synchronous bit streams which carry telephone voice traffic, and
8 b) asynchronous packetized data from LAN traffic.

9

10 Traffic is characterized as having the following characteristics (these characteristics are
11 referred to as the 'traffic contract' for a 'flow');

- 12 a) guaranteed bandwidth
13 b) delay
14 c) delay variance

15

16 The programs (or subroutines) represented by blocks 232 to 235 in Figure 3 adapt the
17 available system resources and deliver the requested 'traffic contract'. The functions
18 include:

19

20 Filtering: Filtering removes unwanted data, either present as unwanted flows, or data
21 within a flow. The effect is to reduce the bandwidth required to support a higher priority
22 flow.

23

24 Bandwidth adjustment: The sub routine represented by block 235 performs a separate,
25 orthogonal function: It increases or decreases the available output bandwidth to match
26 the offered load. That is, it dynamically adds or removes additional bandwidth (on the

